Speech
Technology
Center

Sound Cleaner

Real-time noise cancellation and speech enhancement software

User Manual
Operating Instructions

Speech Technology Center
St.Petersburg, Russia

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and speech enhancement software

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Dear Customer,

Thank you for purchasing this product! For optimum performance and safety, please, read these instructions carefully.

Contents

Necessary Information

This manual describes Sound Cleaner ver. 5.x audio signal processing software.

Listed below are the telephone numbers that you may need if any questions or problems regarding the operation of Speech Technology Center products arise:

Tel:	+7 (812) 331-0665
Fax:	+7 (812) 327-9297
Mail:	Russia, 196084, St.Petersburg, P.O. Box 515
Office:	4 Krasutskogo str., St.Petersburg
E-mail:	**info@speechpro.com**
Web-site:	**http://www.speechpro.com**

When requesting assistance, you should have the following information readily available:

- name of the product and version number
- type of the computer and information about its configuration
- name of OS being used and its version number
- precise description of the problem.

Chapter 1

Overview

Sound Cleaner software is designed to process and play audio signals in real-time mode. It may operate in any MS Windows environment[1]. Real-time audio processing makes Sound Cleaner a powerful tool useful for many different tasks, such as noise-cancellation during audio-monitoring and so on. It also drastically increases productivity and adjustment speed of processing tools.

The main tasks accomplished by Sound Cleaner are:

- Input of audio signals into PC memory and saving them on a hard drive;
- Real-time noise-cancellation and improvement of the signal quality;
- Ascertainment of speech fragments in poor quality phonograms.

[1]If common Windows multimedia sound I/O Card (Sound Blaster) is used. In other cases see **Installing sound I/O hardware** in chapter 2 for more information about OS supported.

Delivery set

Sound Cleaner delivery set includes:

- Sound Cleaner software;
- Wave Assistant signal editor software;
- STC Sound I/O device (optional)[1].

Sound Cleaner capabilities

This manual describes Sound Cleaner software. For details on Wave Assistant software and sound I/O devices see respective manuals and/or technical descriptions.

Sound Cleaner software is capable to:

- Input and save audio signals from microphones and line inputs of playback devices;
- Suppress different kinds of noises and compensate various interferences, including

  ▶ stationary and slowly varying additive complex narrowband (polyharmonic) and broadband noises;

  ▶ slowly varying amplitude-frequency interferences (record or transmission channel AFC inconsistency);

  ▶ short pulse interferences;

  ▶ signal jumps;

  ▶ any kinds of additive noises, provided that audio signal is received through two channels;

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[1] Instead of Sound Blaster card or other windows-compatible multimedia device the software may be delivered with STC-H216 external USB-device. If Sound Cleaner is a part of STC Ikar Lab complex, it will use STC-H189 sound I/O card.

- Process signals in real-time mode. Waveform and instant spectrum of a signal before and after the processing are continuously displayed. You may always adjust any of the processing parameters, no matter which kind of filter is used, instantly monitoring and listening to the effect of changes you have made;
- Adjust playback speed smoothly, without affecting sound pitch;
- Play selected phonogram fragment in loop mode;
- Play a phonogram in pseudo stereo mode with adjustable channel delay;
- Optimize amplitude and frequency of a signal providing best possible audio perception;
- Operate as Direct-X plug-in for most common sound editors;
- Enter text using built-in text editor;
- Automatically create text report of signal processing;
- Save processing parameters for later usage.

Technical characteristics

Main technical parameters of Sound Cleaner are given in the table below.

Number of processed channels	1-2
Processed signal format	8 or 16-bit PCM, μ-law, 24-bit float
Input signal format	16-bit PCM
Saved signal format	16-bit PCM
Sampling rate	8-48 kHz (chosen by user)

Audio file types supported	WAV, DAT, SDT
Maximum phonogram duration	Limited only by free HD space, depends on signal sampling rate
Largest possible number of adaptive filter coefficients	8192
Gain adjustment range	From -60 to +60 dB
Duration of fragment played in loop mode	From 0.5 sec. up to whole phonogram
Playback speed coefficient adjustment range	0.7 - 3
Channel delay value range for pseudo stereo playback	0-20 msec.

Chapter 2

Preparing Sound Cleaner for operation

To prepare the software for operation you should:

1. Install sound I/O hardware and its drivers (if necessary).
2. Install Sound Cleaner and Wave Assistant.

System requirements

Sound Cleaner's PC configuration demands are rather high due to real-time audio processing. For proper operation it requires:

- Pentium III/1000 MHz or better CPU;
- At least 25 Mb of free disk space;
- MS Windows 95/98/2000/XP OS (depending on sound I/O device used - see **Installing sound I/O hardware**);
- At least 128 Mb or more RAM;

- One free USB or PCI slot (depending on sound I/O device used);
- CD drive;
- Color monitor with 1024x768 dpi or better resolution;
- Keyboard, mouse.

During the operation, especially audio input and saving data in a file Sound Cleaner software occupies nearly all the system resources (CPU time, memory, local BUS flow and DMA channels). That's why we advise, that you close all other applications while running Sound Cleaner in order to avoid possible loss of signal fragments.

Important!
Please, note, that Sound Cleaner has a high priority when addressing sound I/O device. This means, that if there is some other application occupying the same device, it may lose access to the I/O device after you shut down Sound Cleaner.

Installing sound I/O hardware

Sound Cleaner may receieve signals from one of three different sound I/O devices: STC-H189, STC-H216 and Sound Blaster. Each of these devices has certain OS limitations, namely, STC-H189 operates only in MS Windows 98, while STC-H216 USB device is supported in MS Windows 2000/XP. Sound Blaster card will work on PC with any MS Windows OS installed.

If you use STC-H216 external USB-device or STC-H189 sound I/O card, you will have to install it as well as necessary drivers according to instructions provided in respective manual and/or device technical description.

Installing the software

After you hardware is properly configured, you should install Sound Cleaner software. To do it place the included CD in your CD-drive and run ***setup.exe*** from **Scleaner** folder on this CD, then follow instructions of the installation wizard. Sound Cleaner and Wave Assistant will be installed on your computer.

> ***Important!***
> If your PC operates under MS Windows 2000 or XP, make sure you have access to the system register (i.e. you have administrator privileges) before you start the installation and registration. You won't need it after the registration is over.

Registration

After the installation the program will run in a demo mode corresponding to Premium version of the software (i.e. all the possible filters and schemes included). You have to complete the registration procedure in order to switch to work mode. To do it run Sound Cleaner from **Programs/Speech Technology Center** group. If you launch the software for the first time, you will be offered to register immediately or do it later, by choosing **Register** from the **Register** menu. In both cases just follow the instructions of registration wizard. You will be asked to enter your **User name** and **Registration code**; both of them are to be found in ***register.inf*** file in **Sclean** folder on the CD. If there is no User name specified, you may enter your own name in this field.

Chapter 3

Basic principles of Sound Cleaner operation

General information

The program operates much like a mosaic - this means you may design a **scheme** of signal processing, combining any of the provided **modules** (**processes**) in a desired sequence starting from sound **Input**. Each process included in the scheme receives a signal form the previous one, applies his own algorithm and then passes the processed signal to the next module in the chain. Number of modules used consequently in a single scheme is limited only by capabilities of your PC. Each module is represented by an icon in the **Scheme window** located in the left part of the Sound Cleaner main window (see Figure 3.1) and has its own dialog window, where all the information and necessary controls are located.

Name of the process is indicated in the header of its dialog window. The header also holds standard **minimize** and **close** (see Figure 3.1) buttons as well as **activate** button. This button is used to apply the module to a signal (**activate** it) or stop using it for signal processing (**deactivate**).

Depending on signal format and type it may be processed along one of the three processing scheme types:

Single-channel schemes are used for mono signals.

Two-channel schemes are designed to process signals in “left” and “right” channel independently.

Stereo filtering scheme is very effective for stereo signals in most cases.

To help you master Sound Cleaner faster and easier we have included a set of standard noise-cancellation schemes in it. These schemes provide good results for typical noises and interferences in most cases.

In case, if Sound Cleaner modules won’t suffice and you will need additional tools, you may wish to employ **Wave Assistant** signal editor supplied with Sound Cleaner. Currently processed signal will be automatically loaded in Wave Assistant if you launch it during processing. The programs are linked so, that if you edit a fragment in Wave Assistant, the changes will be transferred to Sound Cleaner **Input** module, and vice versa. To learn more about Wave Assistant signal editor refer to its manual.

Sound Cleaner main window and process windows

Figure 3.1 displays main window of Sound Cleaner program.

As you see, it is a standard window; its header displays program name and version number. There is a common **menu string** just below the header and a **toolbar**, which contains shortcut buttons to most frequently used menu items. There is also the **messages’ window** at the very bottom, while all the main part of the window is occupied with **process windows**.

As Sound Cleaner loads, you will see all these windows unfold. You may then arrange them as you wish, minimize or hide those, which you

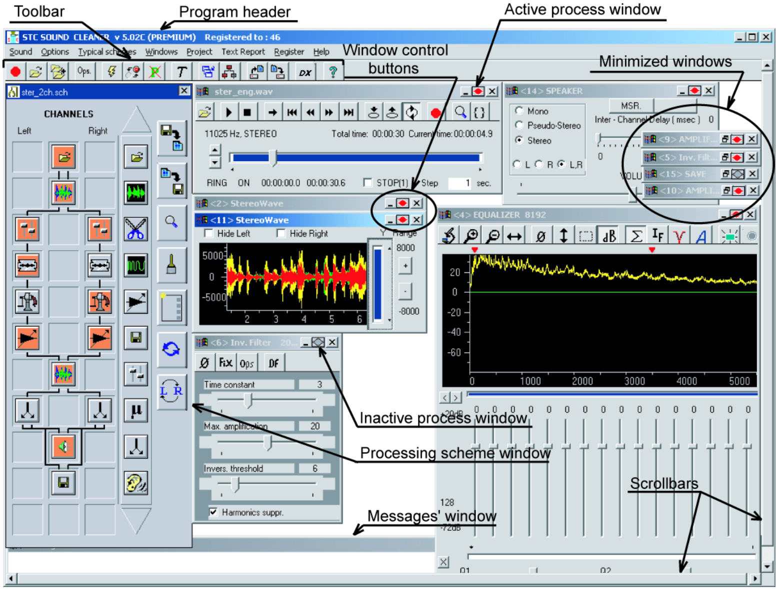


Figure 3.1: Sound Cleaner main window

don't need. You may not, however, change size of these windows except **Equalizer**.

Every window represents a single **process** (**module**) of the currently loaded processing scheme (the only exception is **Processing scheme window** described later in the separate chapter). Note, however, that if a module is included in the scheme, this does not mean, that it is actually used for signal processing, because it may be **inactive**. Each process window header holds additional **Activate** button except common **Minimize** and **Close**: . If this button is greyed, the module is inactive, i.e. is not currently used for processing. To **activate** the process press this button and see it turn red, which indicates, that the process is **active**.

Remember, that all the windows, no matter active or not, may be also

hidden (closed), **minimized** or displayed on the screen. For more details on managing process windows, see description of **Windows** menu below.

Menus

Menu string is located just below the program header. Menu commands control the program in general.

Sound

Sound menu contains commands, which manage input and output of audio data:

Open file. Choosing this menu item will open standard "open file" dialog window. Specify the file you wish to open and process as a signal source.

Save to file. You may choose or create a file to save the processed audio signal into.

Start sound input. Choose this item to start input from sound I/O device.

Play file. This command starts sound input from a specified file.

Stop. Stops sound input.

Pause/Continue. Use this command to pause the playback and then continue audio input.

Global reset. This command restores all the default filter (process) settings and removes all the information accumulated in buffers.

Cancel saving file. This item stops saving data in a file, but does not affect other processes.

Make composite stereo. Choose this command to create a composite stereo - i.e. stereo signal combined from two independent mono signals. See **Processing composite stereo signals** on page 87 for more details.

Switch input channels. Swaps left and right channels of incoming stereo signal.

Options

Commands in this menu set some additional parameters of saving and loading files in Sound Cleaner.

Save. Use this command to specify a file, where current Sound Cleaner configuration will be saved.

Load. There you may choose a file containing saved program configuration and load it.

Details submenu contains three items:

Save file input options when saving to cfg file. If you check this option with the flag, the program will store name of currently loaded audio file and loop settings in configuration file.

Ignore file input options when loading from cfg file. Check this item to ignore audio file saved in configuration file.

Load sound file example together with typical scheme. If this option is checked, Sound Cleaner will automatically load a sound sample associated with particular scheme.

Typical schemes

This menu contains three submenus for managing typical schemes:

Load. . . You may choose a typical processing scheme from a list and load it.

Save as. . . This item saves current scheme and configuration as a typical scheme. See **Typical schemes** section on page 92 for more details on creating typical schemes.

Delete item. . . Choose this command to delete a typical scheme as well as sound sample and configuration file associated with it.

Windows

Commands of this menu manage Sound Cleaner process windows.

Next. Activate next window.

Previous. Activate previous window.

Windows list. Opens **Windows list** dialog window (Figure 3.2).

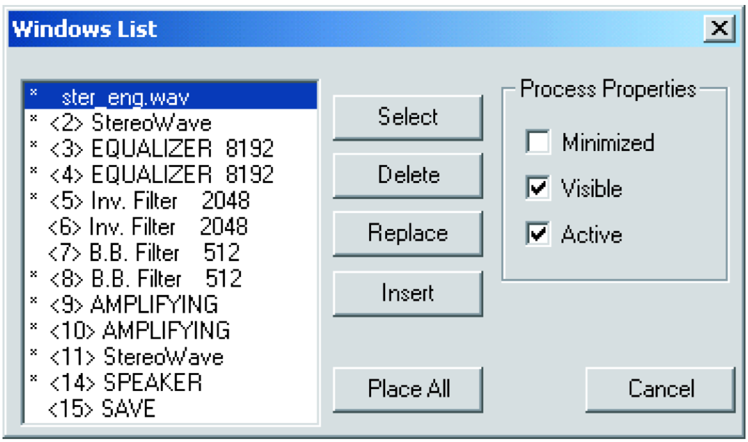


Figure 3.2: Windows list dialog window

In the left half of the window there is a **list of processes** currently included in the scheme. Active modules are marked with an asterisk (*). Left-click on any module to select it. **Process properties** field at the right side will display information about its current status. You may place a flag there to **minimize**, **hide** or **activate** the module.

There are also control buttons in the centre. They perform various operations on the selected process. **Select** button will unfold the window of selected module and place it atop all others (you may do the same if you double-click on any process in the list). **Delete** button will remove the process from the scheme. Pressing **Replace** or **Insert** will open a list of available modules. Double-click on any one of them to place it in the **process list** either instead of the selected one (**Replace**), or before it (**Insert**). Finally, **Place All** button will return all the windows to default position and status.

If you have used any of these commands, the dialog box will be automatically closed. Otherwise press **Cancel** button to return to Sound Cleaner main window.

Place All. Restores default position and status of the process windows.

Hide inactive. This command will hide windows of all currently inactive modules.

Minimize inactive. This item will minimize all windows of currently inactive processes.

Hide minimized. Use this command to hide minimized windows.

Text edit. Opens a built-in text editor window.

View scheme. This item activates sound processing scheme window.

There is also **Options** submenu where you may set default operations with windows. Possible choices are **Hide inactive, Minimize inactive, Hide minimized** and **Windows standard**.

Project

Commands of **Project** menu control and adjust signal processing schemes.

Load scheme. Choose and load existing processing scheme.

Save scheme. Save current scheme in a file specified by user.

View scheme. Activate **Scheme window**.

Load scheme + config, Save scheme + config. These commands load and save processing scheme and configuration file associated with this scheme.

Activate all. Activates all the modules in the currently loaded scheme.

Deactivate all. Makes all the modules in the current scheme inactive.

Text report

There is only one **Create** option in this menu. If you check it, the program will offer you to store processing parameters and current configuration in a text file each time you stop the playback. If you agree, current scheme and configuration will also be saved in the same folder.

Register

There are two available items:

How to register? This command will display information explaining how to register your copy of Sound Cleaner software.

Register. Launches registration wizard (see **Registration** in Chapter 2 for details).

Help

Help topics. This command will open Sound Cleaner help file.

About Sound Cleaner. Brings forth brief information about Sound Cleaner.

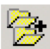
About STC. Displays brief information about Speech Technology Center.

Toolbar

Sound Cleaner toolbar buttons are shortcuts to most frequently used menu commands.


Start audio input from I/O device (same as **Sound/Start sound input**).


Start audio input from a file (**Sound/Open file**).


Open two mono files and combine them into a composite stereo signal (**Sound/Make composite stereo**).

Save/Load program configuration (**Options/Save; Options/Load**).


Restore default parameters for all modules and clear all buffers (**Sound/Global reset**).


Pause/continue signal processing (**Sound/Pause/Continue**).


Start/stop direct listening to source audio signal (without processing).


Open text editor window (**Windows/Text edit**).

 Open **Windows list** window (**Windows/Windows list**).

 Activate scheme window (**Windows/View scheme; Project/View scheme**).

 Load a saved scheme (**Project/Load scheme**).

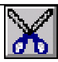
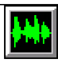
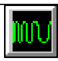
 Save current scheme (**Project/Save scheme**).

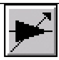
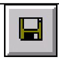
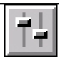
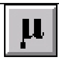
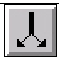
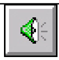
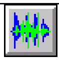
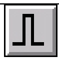
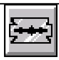
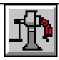
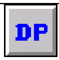
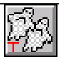
Switch to Direct-X plug-in mode.

 Go to help contents (**Help/Help topics**).

Processes

Sound Cleaner provides consecutive signal processing, i.e. modules are applied to a signal consecutively, one after one. This chain-like sequence is called **processing scheme** and may be saved in ***.sch*** file. Remember, that modules included in the scheme may be inactive. On the other hand, windows of some processes, no matter active or not, may be hidden or minimized. Table below displays names of processes, their icons and short descriptions.

Name	Icon	Description
File input		Input signal from a file
Device input		Input signal from I/O device
Clipping		Clip the signal
Waveform		Signal waveform
Slowing		Changing signal playback speed

Amplifying		Signal amplification
Save		Save signal to file
Equalizer		Parametric equalizer
Mu-transform		Signal transformation (μ-law)
Duplicate		Splitting a signal in two
Speaker		Signal playback
StereoWave		Stereo waveform of a signal
Freq. compensation		Frequency compensation
Impulse filter		Adaptive impulse filtration
Inverse filter		Adaptive inverse filtering
Broadband Filter		Broadband filtering of a signal
Dynamic processing		Signal dynamic processing
Frequency stereo		Frequency stereo filtration
Time stereo		Time stereo filtration

Chapter 4

Audio input

This chapter describes audio **input** process. There are two types of input process depending on signal source: **Input from sound I/O device** and **Input from a file**. Since Sound Cleaner may work only with one signal source at a time, only one kind of audio input may be active and they are united in the single **Input** module. Process window will change according to type of signal source selected. Obviously, **Input** should always be placed first in any processing scheme.

Input from a sound I/O device (ADC)

Main task of this module is to input audio signal from an external sound source. To do it you should connect linear output of playback equipment to linear input of sound I/O device; or, if sound is received from a microphone, connect it to the microphone input of the device.

During the input signal is converted into digital form and passed directly to Sound Cleaner. Processed signal is transferred to linear output of the I/O device and may be at the same time saved via **Save in a file** process. But much more effective way is to save the unprocessed signal on a hard drive or any other media (to do it, activate only **Input from**

sound I/O device and **Save in a file**), and then process it.

Figure 4.1 shows process window. Toolbar below the header contains standard input control buttons: **Start input, Pause** and **Stop input**. **Read sound file** button at the right side of the toolbar will switch to **Input from a file** (see **Input from a file**).

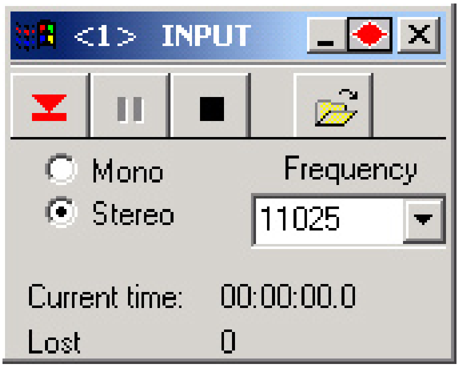


Figure 4.1: Input from sound I/O device window

Use radio button to select **Mono** or **Stereo** input format. Remember, that you have to create or load a scheme for two independent channels to process **stereo signal**.

Frequency droplist contains possible values of **sampling rate** (for STC-H189 you may enter any value within 8-48 kHz range). Default value is 11025 Hz.

Important!
The higher you set input sampling rate, the larger your sound file will be. At 11025 Hz one second of audio takes 22 Kbytes of disk space, while at 22050 Hz – 44 Kbytes. In case of stereo input necessary space is doubled.

Current time counter indicates duration of audio fragment currently being inputted, while **Lost** counter below shows overall length of fragments omitted during current input and/or ADC. If you encounter audio losses, please, consult **Measure mode** in chapter 5.

Input from a file

This process opens specified audio file, passes the information acquired from it to the next active module and controls navigation within a phonogram. Sound Cleaner may input sound from DAT and WAV files. DAT is an extension of audio files used by SIS software and Ikar Lab by STC. WAV file formats supported are:

- 16-bit PCM;
- 8-bit PCM;
- μ- or A-law compression (8-bit);
- IEEE FLOAT - 24-bit.

Input from a file window is shown in Figure 4.2. Slider bar in the center of the window displays current playback and input position. Just below the header is the toolbar. Loop controls and indicators are at the bottom of the window.

Some of the toolbar buttons are just common input and playback controls: **Open file, Start, Stop, Rewind, Fast Forward.**

Other tools control specific input options. After pressing **Go to file time** you may enter time (hh:mm:ss) from the beginning of a phonogram to start input and playback from. Current input position slider will be placed accordingly. You may also drag the slider with your mouse pointer or arrow keys.

You may select to play specific phonogram fragment in a cyclic **loop** mode. To set beginning and end of the loop (**ring**), you may either place

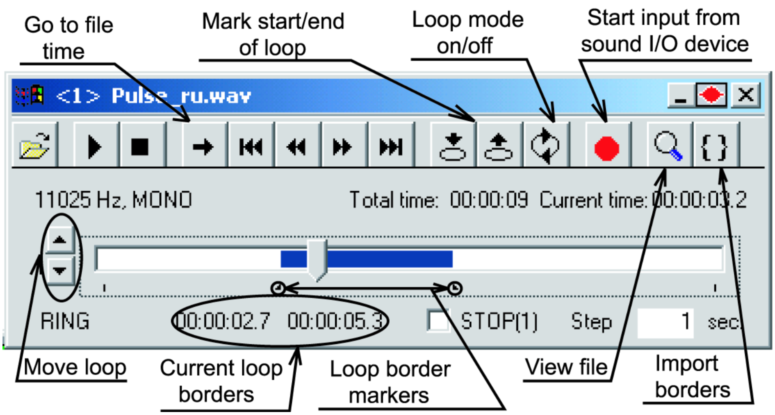


Figure 4.2: Input from sound file window

the slider in a desired location and press **Mark start (end) of loop** button, or simply click left or right mouse button (for left and right border, respectively) over the indicator bar in the middle. Position of both ring borders will be indicated by black **markers** under the bar, while whole looped fragment will be highlighted with blue. At the same time **Current loop borders** digital indicators will display border positions from the beginning of a phonogram (as hh:mm:ss). Press **Loop mode on/off** button to turn this mode on and off. When it is active you should see **ON** message appear at the bottom of the window, next to **RING**.

Start input from sound I/O device button will switch the window to input from a device (see **Input from a sound I/O device (ADC)**).

View file button will launch **Wave Assistant** signal editor software loading currently processed file and loop borders automatically. **Import borders** button will do the opposite, namely, set currently selected in **Wave Assistant** borders as loop borders in Sound Cleaner **Input from a file** process window. See **Wave Assistant** manual for more information on this software.

Sampling rate, mono/stereo **input mode**, **total duration** and **current input position** are indicated below the toolbar.

To the left of the input indicator bar you may see **Move loop** arrow buttons. Press them to move both borders of the ring in any direction with a **Step** specified in the field in the bottom-right corner of the window (in seconds).

Finally, **STOP** flag will, if it is set, enable Sound Cleaner to play the "looped" fragment only once and then stop the input instead of doing it over and over again (as loop-mode is supposed to work).

Chapter 5

Audio output

As with input there are actually two types of audio output available in Sound Cleaner: **playback** (i.e. output to sound I/O device) and **save** (output to file).

Playback

This process (also called **Speaker**) allows you to listen to either processed or source signal, as it is being inputted. So, if you wish to listen to a signal while it is processed, ensure, that **Speaker** module is in the scheme and active.

Playback window is shown in Figure 5.1.

You may choose to play the sound as **mono**, **pseudo stereo** or **stereo**. In **stereo** mode you may also specify with a radio-button, which channel you wish to listen to: **L** means only left channel, **R** – only right one, and **L,R** – both. Do not forget to choose two-channel processing scheme if you are going to play a signal as **stereo** (you may choose ***stereo.sch*** standard scheme). For stereo and pseudo stereo you may also set the inter-channel **delay** within 0-20 msec. range. To adjust **delay** value, move the slider at the right side of the window.

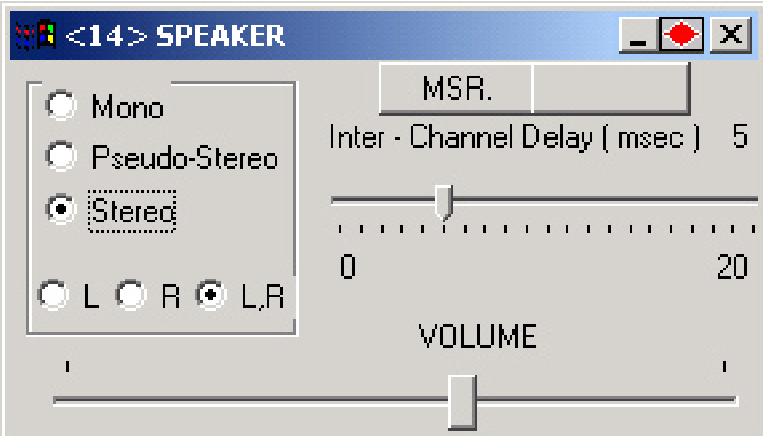


Figure 5.1: Playback process window

Playback **Volume** slider is located in the bottom part. It is linked to MS Windows audio mixer. Try to avoid setting huge volume level, because this will usually lead to considerable signal distortions; it is much more reasonable to adjust signal level with **Amplifier** (see **chapter 6**).

There is also **MSR.** button above the **Delay** slider. Use it to activate **Measure** mode and calculate current PC load.

Pseudo stereo mode

In pseudo stereo mode a signal is played in one of two channels, and then the same signal is played in the other one with a little delay. This makes audio perception more comfortable and increases signal intelligibility; moreover, listeners tend to tire slower when pseudo stereo mode is used. It is especially useful for determining text contents of very long phonograms. In such case you should also try use different processing types and change signal level from time to time.

If you wish to find more useful information about the properties of

human hearing important for perception of noisy speech phonograms, as well as about main principles of determining contents of phonograms, refer to STC **Noise Cancellation and Text Decoding of Low Quality Speech Recordings: Practical Approach**[1].

Measure mode

As you turn on **measure mode** audio playback is paused and automatically calculated **V** value is displayed next to **MSR.** button. This value represents time spent for signal processing compared to total phonogram duration (in percents). **V** monitors PC load, if its value exceeds 100%, it means, that your PC can not process the signal in real time, which, in turn, may result in phonogram fragments being omitted. If you click on the displayed value, calculation will be started over from this particular instant.

To decrease PC load and minimize audio loss probability (if necessary) you should try to:

- Deactivate modules, which provide visual presentation of a signal (**Waveform, Equalizer**);
- Decrease number of bands in **Equalizer** and frame size for adaptive filters;
- Decrease sampling rate;
- Use more fast and productive PC.

Saving signal to analog media

To save processed signal or its fragment on an audio tape or other analog medium, you should first connect record device (tape recorder) line

[1]This brochure is delivered with STC Ikar Lab. It is also available for other Sound Cleaner users – please, contact us for details.

input to line output of your I/O device. Then adjust all the processing parameters and signal level. Finally, you should simultaneously press **Start input** button and record button of a tape recorder. Record will go on until you stop or pause it.

Save to file

To record processed signal in a file you should use **Save** module. It may be saved as WAV or SIS file in mono or stereo 16-bit PCM format, with a sampling rate previously used for signal input. Stereo signals are saved as WAV files only.

Toolbar of the **Save** process window (Figure 5.2) contains common record control buttons.

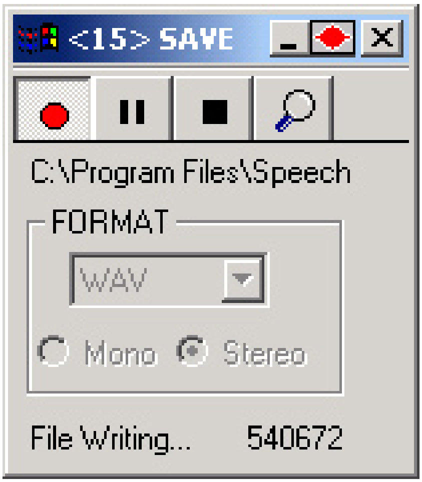


Figure 5.2: Save process window

Save sound to file button opens a dialog window, where you choose file and path to save a phonogram into. It will be displayed just below

the toolbar. After you have specified target file, start the input and the signal will be saved. To temporarily stop saving signal, press **Pause** button. **Close file** button will close the file, so you won't be able to continue recording into it. Finally, you may view and edit this file with **Wave Assistant** by pressing **Export file to Wave Assistant**.

Chapter 6

Amplifier

Amplifier enables you to increase or decrease signal level up to 60 dB. Its window is shown in Figure 6.1.

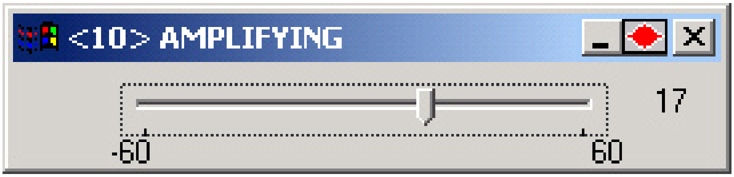


Figure 6.1: Amplifier process window

Gain coefficient is adjusted by the slider with 1 dB step; current value is displayed at the right side of the bar. 0 dB value means, that signal is not amplified.

During signal processing its level may go down due to corrections of frequency characteristic and noise reduction. It weakens some components of a signal which may be then lost during further processing. That's why it is reasonable to amplify a signal during audio processing.

For most effective amplifying **Waveform** module is very useful. We advise that you place it in your scheme right after the **Amplifier** to con-

trol the gain. If waveform of your signal nearly fills the waveform window in Y-direction, the gain is set correctly. Too low gain may lead to gaps in useful signal; while over-amplified signal will be distorted during playback.

If the gain value you set is too high, **overflow indicator** () will appear to the left of the slider bar. Click left mouse button over this indicator to remove it.

Note, that over-amplifying will result in discomfort during playback and in some cases may even cause ear injury, especially if a listener uses headphones for signal playback. So you should avoid making significant gain adjustments instantly. Also remember, that deactivation of processing modules may sometimes greatly increase signal level. For such cases there is **Dependent** option in **Amplifier's** system menu[1]. **Dependent** amplifier will become active only if the process, which stays in the scheme directly before the amplifier, is active. Otherwise amplifier will be automatically rendered inactive.

[1]You may open it by clicking on Microsoft icon in the window's header.

Chapter 7

Waveform

This process is designed to view signals in waveform and monitor their level. It is usually employed in a scheme at least twice: first time to view just inputted signal, and then after all the processing to control it before playback.

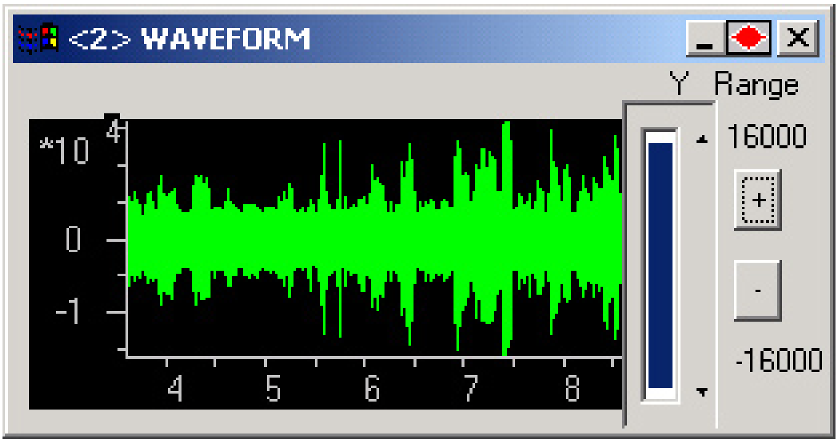


Figure 7.1: Waveform process window

On Figure 7.1 you may see a waveform of a signal displayed in the process window. Current time from phonogram start is displayed along the X-axis, while Y represents signal amplitude in samples (within 20-33000 range). To zoom Y-axis in and out there is a **scalebar** to the right and **+** and **-** buttons. Blue area of the scale represents currently displayed amplitude range. You may set new upper and lower border by clicking on the bar with left or right mouse button respectively. Current borders are indicated with black triangle **markers** at the right side of the bar and their **numerical expressions**.

X-axis is not scalable, i.e. this process always displays **instant waveform** of a signal. Note, that there is also **Stereo waveform** module similar to **Waveform**, the only exception being that the former displays waveforms of signals in both channels.

Chapter 8

Equalizer

This process displays signal spectrum and enables user to correct it via inverse filtering and filter contrasting. **Equalizer** may work in automatic or semi-automatic mode, it is also possible to tune the filter manually to make fine spectrum adjustments. This module may suppress any stationary components of a signal regardless of their frequency and location; it also may be used to raise the amplitude in a chosen spectral band. This filter works well for phonograms containing considerable stationary noises such as power-line noise, mechanical and engine noises and so on.

Number displayed in the window header is current **FFT window size**. The larger it is, the larger is number of equalizer bands and more fine and precise adjustments may be made. To select number of bands you should open **Options** dialog box (described further). Try to set largest possible number of bands to achieve best filtering quality and precision, but remember, that it increases system load as well. FFT window size is strictly determined by number of bands (in fact it is equal to number of bands multiplied by four).

Equalizer controls

Equalizer window may be displayed as **standard** (default) and **large**, the only difference being its size and number of filter adjustment sliders. Figure 8.1 shows standard equalizer window.

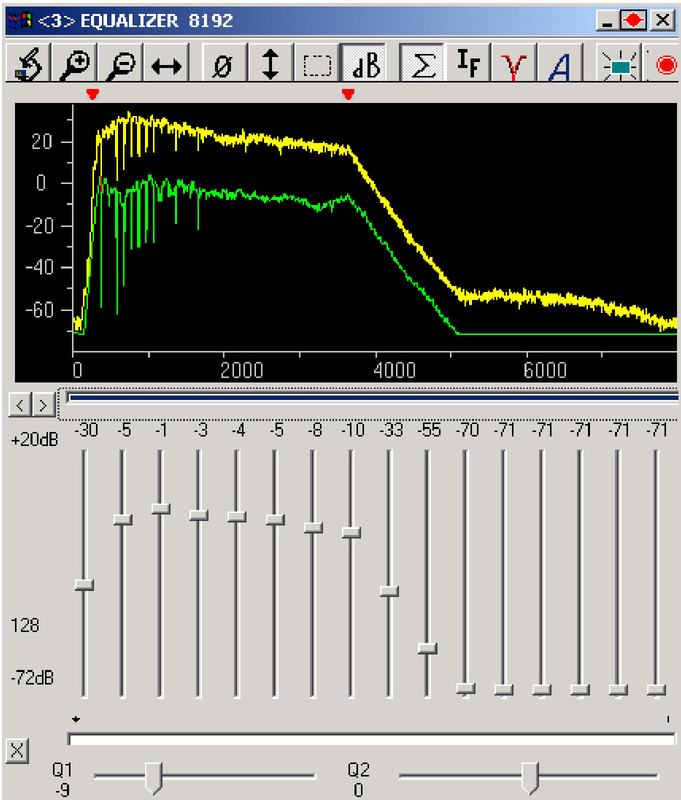


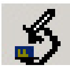
Figure 8.1: Equalizer process window

You can see **toolbar** below the window header and a black window, where current **signal** (yellow) and **filter** (green) spectra are displayed.

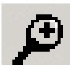
X-axis zoom bar is just under this window; currently visible area is marked with blue. Two red markers at the upper edge of the spectrum window indicate bandpass borders (you may drag them to change the borders). Lower half of the window is occupied with filter band adjustment sliders; **”Elastic”** mode and **Additional FC** controls.

Equalizer toolbar

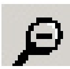
Equalizer toolbar contains most important and frequently used process controls:



Set **maximum horizontal zoom**, i.e. each adjustment slider corresponds to a single filter band.



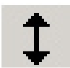
Zoom in, increasing X-axis scale two times.



Zoom out, decreasing X-axis scale two times.

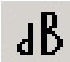
Zoom out, showing **all the spectrum** from 0 Hz up to half of the sampling rate.

Reset filter, placing all the adjustment sliders to zero.

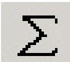


Autoscale Y-axis according to maximum and minimum spectrum values.

Select a **rectangular area** to be displayed in the spectrum window. Select it and press left button and see a dashed rectangle appear, then drag it to specify the area you wish to view, and release the mouse button. Selected area will be enlarged to fit the spectrum window.

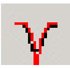


Switch Y-axis from **linear** to **logarithmic (dB)** scale. When this button is pressed, equalizer displays signal in logarithmic scale, otherwise the scale is linear.



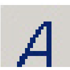
Turn **spectrum accumulation** on and off. While this button is pressed, the program accumulates signal calculating and displaying **average spectrum**. Depress it to switch back to instant spectrum.

Build inverse or harmonic filter (selected by user) basing on current spectrum (either instant or average, if spectrum accumulation is on). Pressing this button automatically stops spectrum accumulation. Note, that either filter is calculated within bandpass borders only.

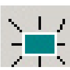


Contrast the filter. See Equalizer options section for more information about filter contrasting.

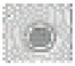
Important!
If you have turned on **filter contrasting** in **Options** menu, it will be automatically done during inverse filter calculation and/or automatic filtration. In this case pressing this button will make Sound Cleaner to contrast the filter once more.



Automatic filtration button turns off spectrum accumulation and then inverse or harmonic filter calculation (filter type is selected by user). You may set time of automatic spectrum accumulation in **Options** menu.



Switch between **standard** and **large equalizer window**.



This indicator button turns red if a signal was over-amplified which caused an **overflow** during equalizer output. Press the button to bring the indicator back to passive state until the next overflow.

Equalizer options

If you click your left mouse button over Microsoft sign in the header of equalizer window, its standard system menu will appear. It has, however, additional **Options** entry, that opens **Additional options** dialog window (see Figure 8.2).

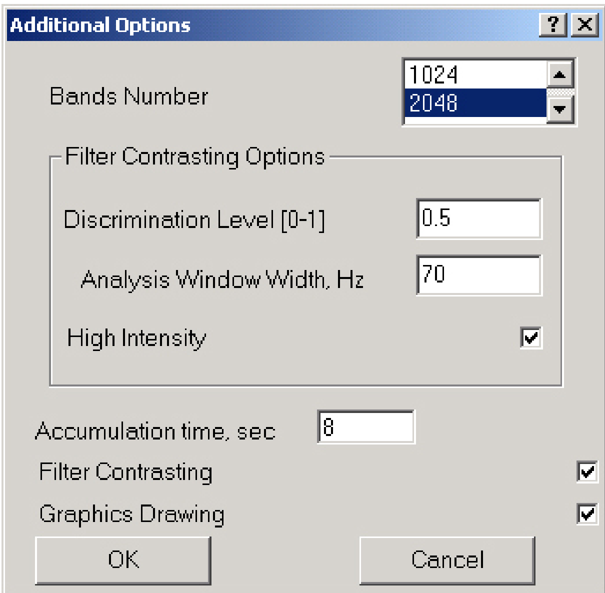


Figure 8.2: Equalizer options window

There you may choose **number of bands** for the equalizer. Gener-

ally speaking, setting greater number of bands will increase filter performance as well as your PC load.

Filter contrasting options field occupies the center of the window. Contrasting means that Sound Cleaner will automatically detect narrow gaps in the filter FC and then broaden and deepen them. This operation may considerably improve filtering quality, especially if there are clear local noise peaks in the spectrum of a signal. To enable **filter contrasting** check the respective flag in the bottom part of the window. Then adjust contrasting options:

- **Discrimination level** parameter represents relation between filter value in the gap and on its edge. The program uses it to determine which filter gaps are to be contrasted. **1** value means, that it will contrast all the local minimums; **0** – filter won't change. Values around **0,5** will make the program to skip small (usually, natural) minimums at the same time contrasting large and distinct gaps.

- **Analysis window width** value determines maximum width of a gap (in Hz), which will be considered ”narrow” and, therefore, will be subject to contrasting. Default value is 70 Hz.

- **High intensity** flag, when checked, will slightly broaden the gaps in addition to common contrasting effect.

At the lower part of the screen you may see a group of additional controls. **Accumulation time** value is a duration of spectrum accumulation for automatic filter calculation. **Filter contrasting** flag turns on respective operation, as was mentioned earlier; **Graphics drawing** may be turned off to disable drawing of signal spectrum, enhancing performance of weak PCs.

OK button will close the window saving all the changes, **Cancel** will discard them.

Zooming and scrolling

To zoom signal spectrum window in and out, you may use appropriate toolbar buttons (see Equalizer toolbar), **“<>”** buttons and horizontal zoom bar located below the spectrum area. Blue-marked fragment of this bar indicates, which part of the whole spectrum is currently displayed. If the whole bar is blue, then all the spectrum is displayed (if, for example, ↔ button was pressed).

Click left or right mouse button over the bar to set respective borders of displayed spectrum. You may also scroll it with arrow buttons to the left of the bar or with arrow keys. In the latter case you have to set the focus on the bar. Pressing the button or key moves the displayed area 1/16 (1/32 for large window) part of the whole signal area to the right or left.

Adjusting filter FC

There are 16 spectrum adjustment sliders in standard window and 32 in the large. Current signal level adjustment value in controlled band is indicated above each slider (in dB). Highest and lowest possible values (+20/-72 dB), which correspond to extreme slider positions, are given near the left edge of the window. There you may also see a **number of spectral bands**, which are currently controlled by a single slider. If you zoom in and out, this number will change, reaching 1 at maximum X-axis zoom level.

“Elastic” mode

“Elastic” mode enables you to simultaneously adjust several sliders as if they were bound together with an elastic thread. In this mode it is much easier to change filter FC smoothly.

To include sliders in an “elastic” group, you should click left and right mouse button on the bar below the sliders, thus setting left and right border of the group. Selected group will be marked with blue indicator in the bar.

Note, that only sliders (not signal bands!) may be grouped and bound together. This means, that if you zoom in or out, those sliders, which you have previously included in a group will control other signal bands.

x button to the left of the “elastic” mode bar turns **linearization (smoothing)** on and off. Outlines of a filter built in “elastic” mode will be smooth (not jagged) if this mode is on.

Additional FC adjustment

Sliders **Q1** and **Q2** located at the very bottom of the screen provide additional filter FC adjustment, which are added to values set by FC adjustment sliders. This extra adjustment makes speech sound more natural and be more comfortable for the listener’s perception.

Q1 adjusts FC convexity within 100-800 Hz frequency band. **Q2** changes FC increase/decrease for every 1000 Hz starting from 1000 Hz. Both sliders work within -18/+18 dB range with current value indicated to the left of it.

Chapter 9

Adaptive broadband filtering

Adaptive broadband filter is based upon **adaptive frequency algorithm**. This algorithm is designed to suppress broadband and periodic noises due to electric pick-ups or mechanic vibrations, room and street noise, communication channel or record equipment interferences. You may hear these noises as hum, rumbling, hisses or roars. Note, that this algorithm performs at SNR not worse than -5 Hz. It is nearly impossible to remove such noises with other methods, such as one-channel adaptive filtration, spectrum smoothing or equalizer, because they are spread across the whole spectrum and intersect with speech signal.

Broadband filtering controls

Broadband filter window is shown in Figure 9.1.

There are four process control buttons located on the toolbar:

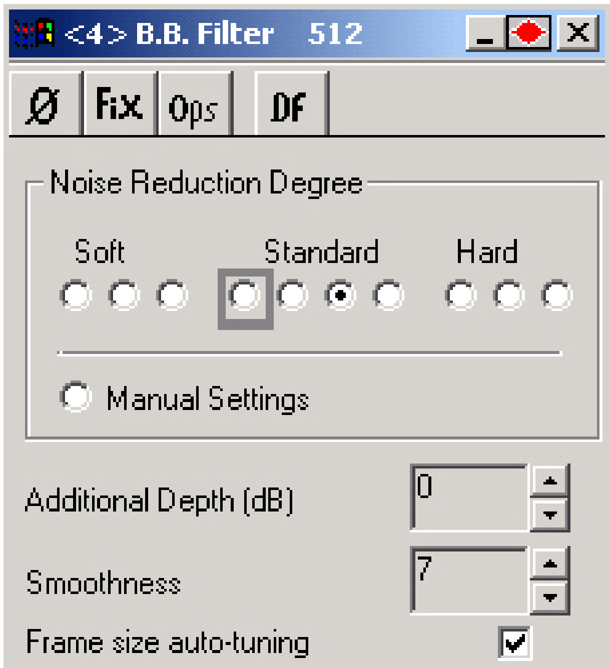
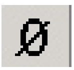
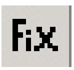


Figure 9.1: Broadband filter process window



Reset filter



Fix current filter settings

Open **Auxiliary options** dialog window

Restore **default** filter settings

Noise reduction degree group enables you to either specify level of **automatic** filtering, or use **Manual settings** for more subtle filter adjust-

ment. **Additional depth** (available only in automatic mode), **Smoothness** and **Frame size auto-tuning** fields will be described later.

Automatic mode

Automatic noise filtering works best if you have to get reliable results very fast. It also does not demand from the user any special noise-filtering skills and experience.

All you have to do in this mode is specify desired noise-reduction level (soft, standard or hard). Each level has also its own inner scale, so in total you have 10 grades of noise-cancellation. Just choose appropriate radio-button and the program will automatically select optimum values of noise suppression **Intensity and Depth**. Note, that filter needs some time to tune itself to a specific noise, so you should try to avoid making considerable instant changes of noise-reduction level.

You may also choose **Additional depth** of noise-reduction within -15/+15 dB range and enable **Frame size auto-tuning**. These settings will be described later in this chapter.

Manual mode

For experienced users, who are not completely satisfied with results of automatic processing, there is a possibility to tune all the necessary parameters manually. To do it just select **Manual settings** in **Noise-reduction** degree group. This will automatically open **Auxiliary options** window (Figure 9.2).

Main and most important parameters of broadband filtering are suppression **Intensity** and **Depth** located in **Manual settings** group.

Suppression intensity adjusts inflection point of SNR/suppression curve (1-40). **Suppression depth** may be set within 1-80 range; it determines largest possible suppression of spectral components of a signal.

Generally speaking, increasing both these values will lead to better noise-reduction, but useful signal may also be suppressed and speech

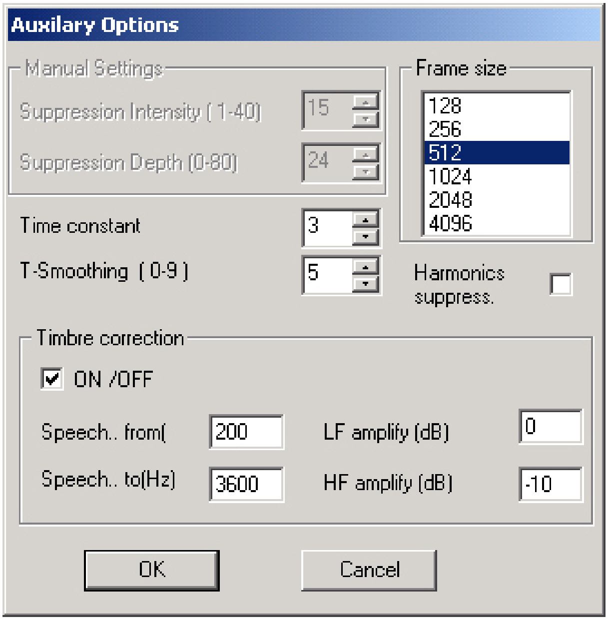


Figure 9.2: BB filter auxiliary options window

quality and intelligibility reduced. We advise, that you adjust these parameters very carefully and always control achieved effects.

Time constant field sets the time of filter adjustment to signal spectrum variations. In most common cases 3-4 sec. is recommended, for non stationary noises the value should be slightly decreased (to 1-2 seconds). Setting less than 1 second **time constant** will, in most cases, greatly decrease speech quality.

T-smoothing parameter controls time smoothing of filter coefficients. It usually helps to remove musical noises which sometimes appear after broadband processing. For large SNR values and signal sampling rate less, than 11025 0-1 values should suffice, while bad SNR and huge sampling rates may demand T-smoothing of approximately 5. Default

value is 1.

In the **Frame size** list you may choose spectral resolution, i.e. number of spectral bands and size of a signal processed audio block. As this value is increased, more different sounds are mixed together in a single block, so there are fewer spectral bands, containing no speech signal. Larger frame size tends to produce considerable echo effect. On the other side, smaller values lead to weaker and less precise noise-reduction. Available **Frame size** range is 128-4096 with 512 being the default value. In fact, it depends on sampling rate of a signal, but you may check **Frame size auto-tuning** in the filter main window to make Sound Cleaner to choose frame size automatically.

Harmonic suppression flag turns on additional suppression of harmonic noises.

Timbre correction is necessary to remove unnecessary low- and high-frequency components of a signal, i.e. configure the bandpass so, that it would suit human speech frequency range. In this case speech becomes more intelligible and comfortable to listen to. You may turn timbre correction on and off with **ON/OFF** flag and adjust its parameters.

Speech . . . from sets bandpass lower border and **Speech . . . to** – the upper border. For 10-11 kHz sampling rate 200 Hz lower and 3600 Hz upper borders are recommended. In fact, these values depend on signal quality and sampling rate. As a rule you should somewhat lift the upper border value. If you encounter strong broadband noise (hisses, rumbles), it is reasonable to decrease this value to 2900-3200 Hz.

HF and **LF amplify** parameters adjust sound level in high- or low-frequency area. They work exactly the same as **Q1** and **Q2** sliders of the equalizer (see **Additional FC adjustment** of the previous chapter). For common cases it may be useful to reduce high frequencies, setting **HF amplify** value to approximately -3/-6 dB. This is not a rule, however, in fact, you may have to leave it unchanged or even raise this value.

There are also two parameters of the main window, which are not

yet described. **Smoothness** defines frequency smoothing of filter coefficients which (like T-smoothing) is useful for suppressing musical tones, which sometimes appear after processing.

Frame size auto-tuning makes Sound Cleaner automatically choose the frame size according to sampling rate of the signal. If you try to do it manually, this flag will be automatically removed.

Model filtration

If a noise you are going to remove with the broadband filter is stationary or close to it (i.e. spectral envelope does not change in time), and a phonogram contains a fragment of this noise without useful signal (this fragment should be long enough for the filter to adjust itself), then you may use this fragment as a **model** for filtration. To do it find such fragment (**Wave assistant** may be very helpful) and run it in the loop mode. Adjust the broadband filter so, that it would completely suppress the signal (noise in this case), then press **"Fix"** button on the toolbar. Filter coefficients will be set and used for filtering the entire phonogram without any adjustment.

Chapter 10

Adaptive inverse filtration

Adaptive inverse filtration process is based upon **Adaptive Spectral Correction** algorithm, sometimes called also **Adaptive Spectral Smoothing**.

Adaptive inverse filtration effectively suppresses strong periodic noises from electrical pick-ups or mechanical vibrations thus recovering speech signal and equalizing signal AFC. It amplifies weaker signal components and suppresses the stronger ones at the same time. The average spectrum therefore tends to approach the flat spectrum unmasking the speech signal and improving its intelligibility. Broadband noises, however, usually become stronger making signal perception less comfortable. It means that you should try to reach a compromise between noise reduction and speech perception.

Adaptive inverse filtering controls

Inverse filtration window is shown in Figure 10.1. Its toolbar contains four common filter control buttons:

Reset filter

Fix current filter settings

Open **Auxiliary options** dialog window

Restore **default** filter settings

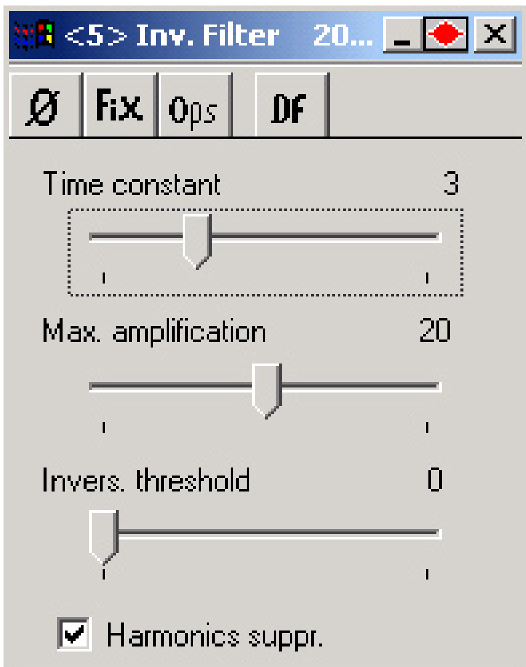


Figure 10.1: Adaptive inverse filter window

Time constant slider sets the time which the filter needs to adjust to variations of the spectrum within 0,1–10 seconds range. In common

cases 3-4 sec. values are recommended; for non stationary noises decreasing this value to 1-2 seconds may be useful.

Max. amplification limits weak spectrum components gain for any given frequency. It is necessary to avoid raising the level of noise during the pauses. 20-30 dB values are recommended.

Inversion threshold value marks certain level of the signal which is considered the border between "weak" and "strong", i.e. all the components below that level are amplified, while those, that exceed it are weakened.

Ops. toolbar button opens **Auxiliary options** dialog window (Figure 10.2). You may choose **frame size** from the list box and configure **Timbre correction** parameters. Timbre correction settings are described in **Manual mode** section of **Adaptive broadband filtering**.

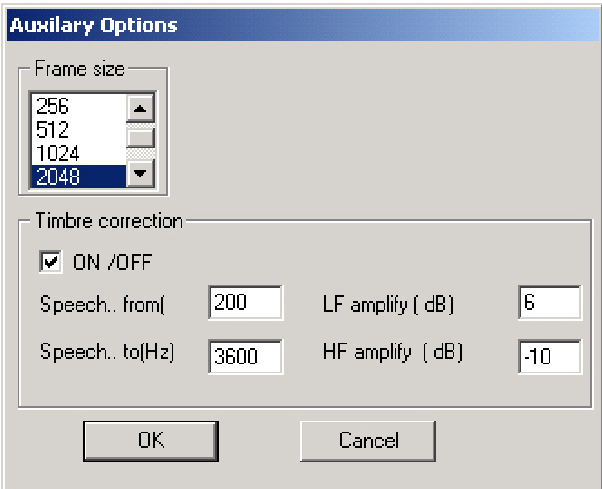


Figure 10.2: Adaptive inverse filter auxiliary options window

Chapter 11

Frequency compensation

Adaptive compensation process uses Widrow **adaptive filtering algorithm** for **one-channel adaptive compensation**. It is most effective for narrow-band stationary and regular interferences. The filter adjusts itself smoothly maintaining good useful signal (speech) quality. Except frequency compensation this process also provides adaptive compensation in time domain. It also suppresses harmonic interferences provided their phase is more or less stable.

Adaptive compensation algorithm

Adaptive compensation enables user to remove both narrowband stationary interferences as well as regular ones (vibrations, power-line pickups, electrical device noises, steady music, room, car and water noises, reverberation and so on). One-channel adaptive frequency compensation recovers the speech suppressing tonal interference by 20–40 dB.

Main advantage of this process is its capability to preserve the speech signal much better than other filters usually do. It happens because the interference is in this case subtracted from a signal and not multiplied by 0. In some cases only a part of periodic interference may be removed,

so you may use adaptive compensation in your scheme more than once.

Primary compensation parameters, which affect noise suppression level, are **the number of filter coefficients** (defined by **frame size**) and **delay** value. Increasing number of coefficient allows to suppress more spectral interference peaks at the same time lowering filter adjustment speed. Delay should not be set lower than half the number of filter coefficients.

Frequency compensation controls

Figure 11.1 displays **Frequency compensation** window.

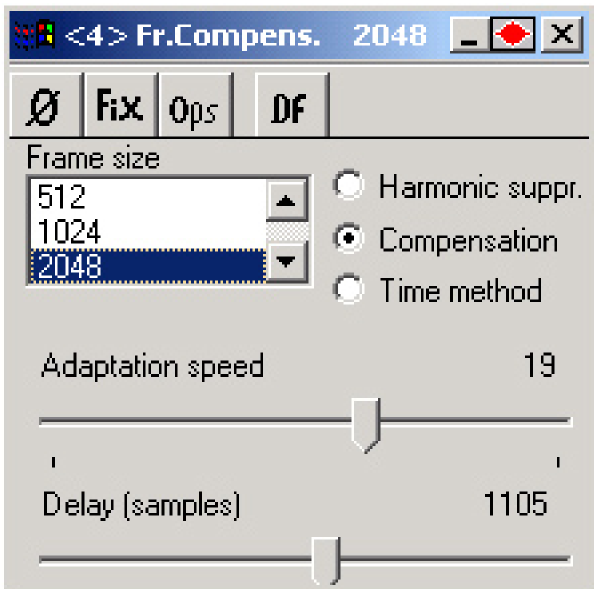


Figure 11.1: Frequency compensation window

Frequency compensation **toolbar buttons** perform common func-

tions: **Reset filter, Fix filter, Auxiliary options** and **Restore default filter settings**.

To choose compensation mode use radio-buttons at the right side of the screen:

Harmonic suppression allows to remove harmonic interferences provided its phase is relatively stable.

Compensation is used to suppress interferences in frequency domain.

Time method activates compensation in time domain.

Frame size is the most important compensation parameter in any mode. It defines number of spectral bands and processed data block size. The larger it is, the more **filter coefficients** is used for compensation, and, therefore, more spectral peaks may be removed. In common cases greater **frame size** will lead to better compensation, though it may also cause **echo** effects. 512-2048 samples should be sufficient for most phonograms.

Adaptation speed slider sets filter **adjustment time**, i.e. time the filter needs to tune itself to the variations of interference spectrum. For common cases values around 20-25 are recommended. If an interference spectral parameters change quickly you should try to increase **adaptation speed** and vice versa. Remember, that large filter adjustment speed tends to impair speech signal quality.

Delay may be set for **time** (0–1000 msec.) and **frequency** (from 0 to **frame size**) compensation. This slider sets an interval between the fragment, where the interference is calculated and the beginning of compensation. For frequency compensation delay should not be less than one half of the chosen frame size; for time compensation – 25 msec. or more is recommended. Otherwise speech signal may be greatly distorted.

Frequency compensation **Auxiliary options** window (Figure 11.2) is opened with **Ops.** toolbar button. It contains **Timbre correction** settings (see **chapter 9**, **Manual mode**) and **Adaptation threshold** slider.

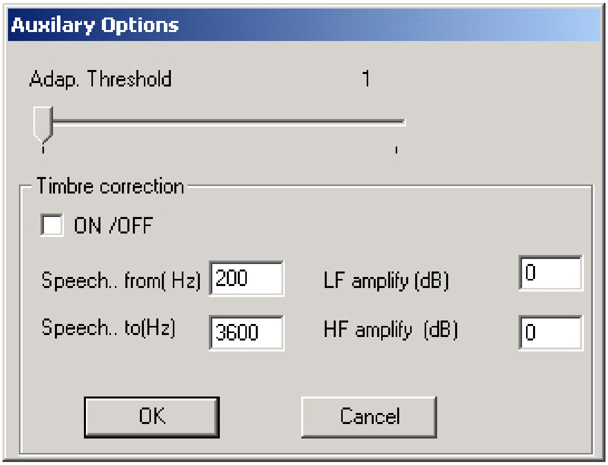


Figure 11.2: Frequency compensation auxiliary options window

Adaptation threshold may vary within 1-32000 dB. If signal amplitude exceeds this value, necessary filter adjustments are made. Thus, you may set the threshold to keep filter unchanged during low signal level phonogram fragments, for example, when there is no interference.

Chapter 12

Slowing

This process is used to **increase** or **decrease signal playback speed** without changing the voice pitch. The window (Figure 12.1) contains only playback speed adjustment slider. Current speed coefficient (in %) is displayed at the right side.

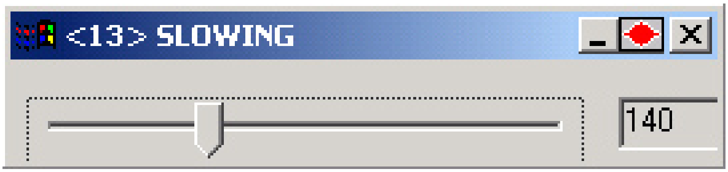


Figure 12.1: Slowing process window

Speed adjustment is usually employed in the very end of the scheme, just before the playback of processed sound.

Chapter 13

Clipping

This process is necessary to limit amplitude of a signal, thus removing strong bursts and smoothing the signal's level. On the other hand it does not significantly affect speech intelligibility. **Clipping** is also very useful for rough clearing of harmonic interferences, provided there are rather long fragments of interference, and its level exceeds the level of useful signal.

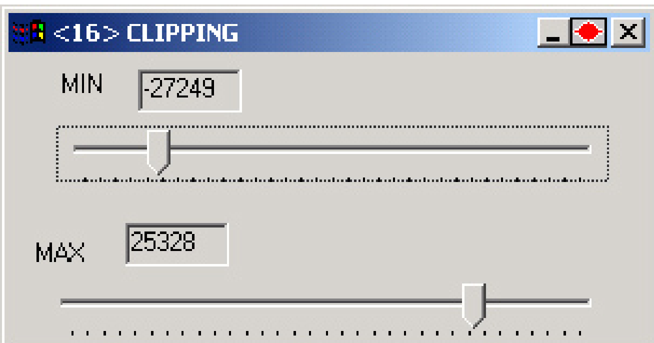


Figure 13.1: Clipping process window

To clip the signal you just have to point **minimum** and **maximum** signal level values using **MIN** and **MAX** sliders in the process window (Figure 13.1). Current value is shown in a field above the slider.

Then all the values of signal level, which are higher than maximum and lower than minimum value are discarded and made equal to respective value.

Chapter 14

Mu-transformation

Mu-transformation is recommended if signal fragments greatly differ in level. It happens, for example, if one of the speakers is located near the microphone and the other one - quite far from it. During the transformation weak fragments are greatly amplified, while strong ones are amplified insignificantly or not amplified at all. This process is in fact similar to **dynamic processing** described later in **chapter 16** but works faster.

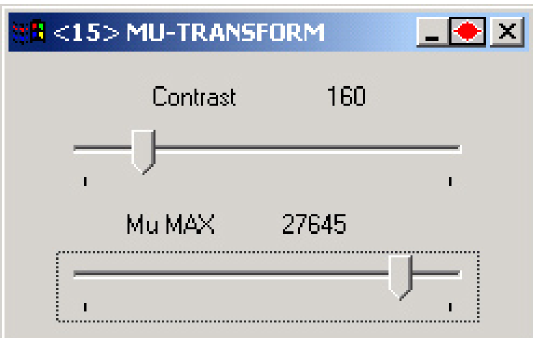


Figure 14.1: Mu-transformation process window

There are two parameters of **mu-transformation** (see Figure 14.1).

Mu MAX defines highest possible level of signal after the transformation (1-32000).

Contrast value sets the number of mu-transformation coefficients and defines signal level range to be transformed with maximum amplification. **1** value means that all the input signal will be evenly amplified and brought to maximum during transformation. **1000** value will, on the contrary, greatly amplify the weak signals, reducing at the same time the level of strong ones.

Chapter 15

Impulse filtration

Adaptive impulse filtering automatically restores speech or musical fragments distorted and masked by various **pulse interferences** such as clicks, radio noises, knocks and so on. Adaptive impulse filtering algorithms improve quality of the signal suppressing powerful signal impulses and thus unmasking useful audio signal and increasing its intelligibility.

Impulse filtration algorithm

During impulse filtration Sound Cleaner substitutes impulses with smoothened and weakened interpolated signals. If the program does not detect an impulse, it leaves the fragment unchanged. It also does not suppress tonal interferences and broadband noises. Impulse detection is based upon the information, which the program has about differences between useful signal and an interference. Thus, setting filtration parameters correctly is critical for effective processing.

Impulse filtration controls

Impulse filter window is shown in Figure 15.1.

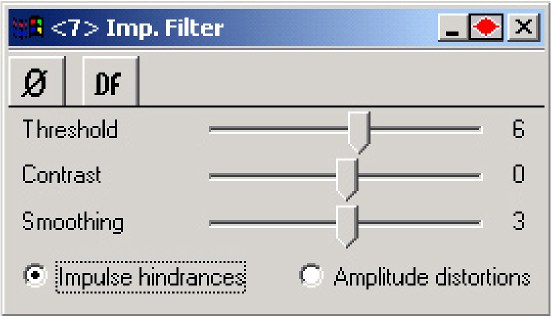


Figure 15.1: Impulse filtration process window

The process may work in two modes, switched by the radio-button at the bottom edge of the window:

Impulse hindrances – in this mode short (up to 100 samples) impulses will be smoothened and removed, being substituted with values interpolated from the useful signal.

Amplitude distortions works for the the same short impulses as well as for distortions due to overload (if, for example, ADC limitations were violated during audio input). In this mode the distortions are smoothened, so you will not have to adjust any processing parameters.

Impulse filtration is configured with three sliders:

Threshold is necessary to detect and locate the impulses basing upon their energy. Default value is 6. Decreasing detection threshold will make the program locate weaker impulses. Setting higher threshold values will leave the weak impulses intact; if the value you have set is too small, then sudden changes of useful signal level will be considered impulse interferences and removed.

Smoothing slider controls detection and localization of impulses

with respect to their duration. Decreasing this coefficient will enable the program to detect short and weak impulses, impairing, however, localization of longer impulses.

Contrast works approximately the same as **threshold**, but performs more subtle filter tuning. Generally, increasing the contrast will enable Sound Cleaner to detect more impulses. It is usually adjusted in the end, after appropriate **threshold** and smoothing values are selected.

Except these settings impulse filter is also supplied with common **Reset filter** and **Restore default filter settings** buttons located on the **toolbar**.

Chapter 16

Dynamic processing

Dynamic signal processing improves its intelligibility if the signal fragments greatly differ in level, in case of resonant knocks (i.e. long impulses) and room noises. Dynamic processing algorithms improve and unmask the signal suppressing powerful impulses and clicks and reduce the listener's fatigue for long phonograms. It is in some way similar to **mu-transformation** (see **chapter 14**. **Mu-transformation**), but works a little slower. On the other hand dynamic processing has a major advantage over mu-transformation: it brings no distortions into the signal.

Dynamic processing controls

Figure 16.1 displays **dynamic processing window**.

Toolbar contains common **Reset filter** and **Restore default filter settings** buttons. **Threshold** set by the slider in the bottom of the window is very important. It is used to distinguish **strong** and **weak** signals. If you choose to **weaken** the **strong** signals, they will be also brought down exactly to the threshold value; while weak signals will be amplified to one tenth of this threshold.

Radio-buttons in the middle set separate dynamic processing modes

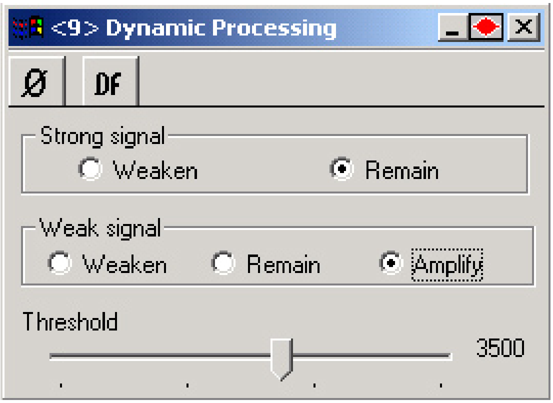


Figure 16.1: Dynamic processing window

for **strong** and **weak** signals. Strong signals may **remain** at their initial level or be **weakened**, which is usually done to bring loud speech down to threshold value or eliminate strong and long (more than 20 sec.) impulses (knocks).

You may also **amplify** weak signals to balance the level of speech for two speakers; leave them unchanged (**remain**); or **weaken** them, which may be useful for suppressing the noise in pauses between loud speech fragments.

Chapter 17

Stereo processing

There are two different kinds of **stereo processing** available in Sound Cleaner: **two-channel signal processing** and **adaptive stereo filtering**. In the first case the sound in each channel is processed independently, while in the second case data acquired from one channel (called **referent**) are used for filtering the signal in the second one (**primary channel**).

Independent two-channel processing

Before you start stereo processing you should load appropriate processing scheme (**Typical schemes/Load.../Standard scheme of 2-channel signal processing**) or create a two-channel scheme yourself. Then you may use any of the described processes for each channel independently.

If you have used different filters or different parameters for these two channels, you will have to include the **Stereo Wave** process into your scheme.

StereoWave

This process is used to display audio data from both processed channels in a single window. It is absolutely similar to **Waveform** (see **chapter 7**). The only exception are the **Hide left/Hide right** checkboxes above the waveform, which allows to hide a waveform of a signal acquired through particular channel.

Adaptive stereo filtering

As in the previous case, you will have to create or load a scheme, where one of the **adaptive stereo filtration** modules is present. There are two appropriate typical schemes (**Typical schemes/Load.../Stereo filtration**): for filtration in **time** and **frequency** domain. The only difference between these two schemes is the type of active stereo filter (using them together for the same signal makes no sense anyway).

Two-channel adaptive filtration algorithms are designed to suppress both unstationary broadband (background speech, radio, room noise) and periodical (vibrations, power-line pick-ups) noises. The backbone of these methods is acquiring the information about the interference from the reference channel and the using it to remove this interference from the primary channel.

Stereo filtering has, however, a serious limitation: it may be effectively applied only if following conditions were met during the audio record:

- Stereoscopic base (i.e. distance between the microphones) should not be less, than 10 cm;

- Primary and reference channel microphones should be located at different distance from the signal source. The same is true for the source of interference;

- Both useful and interfering signal received by microphones should come directly from the source (not be reflected).

Stereo filtering in time domain

Time filtration allows to suppress unstationary broadband noise (speech, music) and periodical interferences (vibrations, power-line pick-ups) from some point-like source (e.g. radio set). Its process window is shown in Figure 17.1.

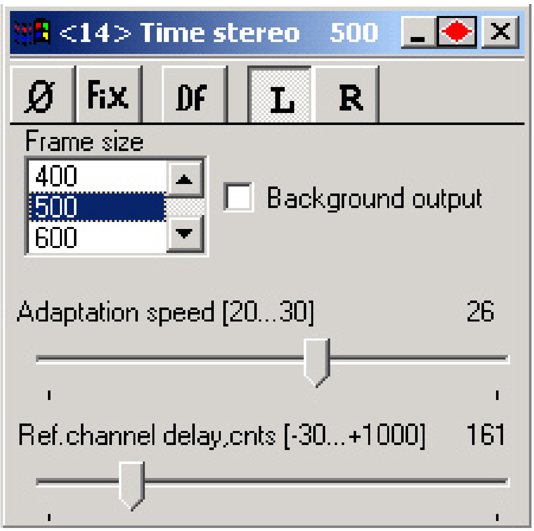


Figure 17.1: Time filtration process window

Toolbar contains common **Reset filter, Fix filter** and **Set default filter values** buttons. There are also two additional **L** and **R** buttons necessary to specify, which channel should be considered primary (L stays for left channel and R - for right).

Frame size should be selected from the list and may vary between 50 and 2000 counts. We recommend you to listen to results after ev-

ery frame size change and then make further adjustments, if necessary. Frame size defines number of filter coefficients; large values usually slow down filter adjustment but provide better filter performance, especially when removing reverberation noises.

Background output is used to monitor and control filter performance. If you place this flag, the program will play back the signal which, with current filter settings, is considered noise and removed. If all the settings are correct, background output should contain only interference signal. Nevertheless, if you hear useful signal in the background, there may be two reasons for it:

1. You may have swapped primary and reference channel. In this case all you have to do is just choose the other channel as primary with **L/R** toolbar buttons.

2. Useful signal is partially removed. It means, that reference channel microphone receives not only interference, but also useful signal. To solve this problem try moving reference channel microphone, if possible, to minimize useful signal received by it.

Adaptation speed slider defines filter auto-adjustment speed. For unstationary noises 26-29 values are recommended, while for slow varying ones 22-25 should suffice.

Reference channel delay is a parameter critical for effective stereo processing. It is set in counts and lies within -30 – 1000 borders. Delay should be chosen with respect to one important rule: primary channel interference should never anticipate the reference channel. For example, if primary and reference signal are simultaneous, setting positive delay will make correct filtering impossible. Introducing, on the contrary, small negative delay will lead to reference signal anticipating the primary, which is necessary for correct filter adjustment and effective processing.

In most cases **delay** should be close to 0 or a little bit less. But if the reference signal is recorded from an interfering device output, setting

huge positive delay may be necessary.

Note, that 1 count delay at 8000 Hz sampling rate is equal to 2,3 cm propagation difference (i.e. difference between distances the signal has to cover in order to reach primary and reference microphone). Thus, 1 m propagation difference will demand about 40 counts **delay** for good processing results.

Stereo filtering in frequency domain

Frequency (spectral) filtration differs from **filtration in time domain** only in processing methods and conditions. **Delay** for the frequency filtration is set in centimeters, not in counts. It represents the difference of distances between microphones and interference source. In all other aspects **delay** has the same meaning and is set the same way as for time filtering. **Frame size, background output** and **adaptation speed** also work exactly as described in previous section.

Frequency stereo filtration window is shown in Figure 17.2.

Process toolbar is similar to time filtration window, the only difference is **Options** button, which opens frequency filtering auxiliary options dialog window. In this window you may configure standard **timbre correction** settings and set maximum **suppression** depth.

There are three available modes of **Frequency filtration**. They are selected with the radio-buttons.

Point source noise mode works approximately the same as time filtration, but with better speed because of larger **frame size** values. On the other hand the filter adjusts itself to an interference a little slower. It is very effective when the reference signal is recorded from a line output of interfering device (e.g. radio set). In this case you should set appropriate positive delay because the source is actually located in some distance.

Reference channel noise mode removes noise of some distance source (working engine, background speech) from the primary chan-

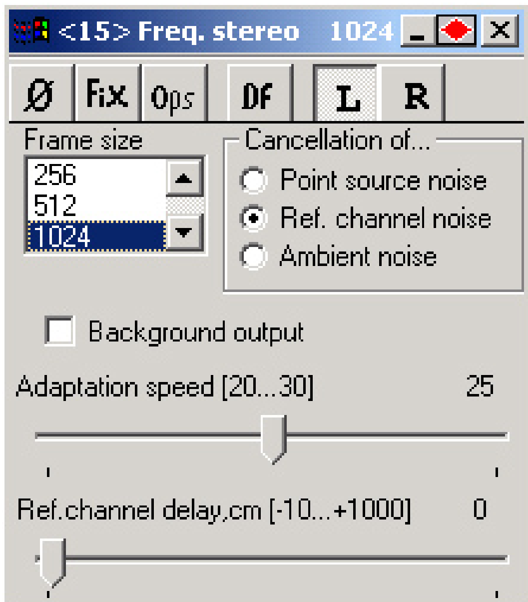


Figure 17.2: Frequency filtration process window

nel provided the reference microphone is set next to this source. You will also have to set an appropriate positive delay.

Ambient noise mode allows to clear useful signal if it is received by two closely (30-70 cm) located microphones from a nearby (less than 1,5 m) source, removing reflected noise of a distant source (air conditioner, orchestra and so on). It works best if the interference source is located symmetrically relative to the pair of microphones. If not, you may correct the symmetry with appropriate delay setting.

Processing composite stereo signals

In Sound Cleaner there is a possibility to create a **composite stereo signal** by combining two different mono signals into one. It may help you to apply **adaptive stereo processing** to signals recorded by two different microphones.

To be merged in a composite stereo mono signals must be of the same type (***wav, dat***) and recorded with the same sampling rate. If these signals have different length, then duration of resulting composite stereo will be equal to the length of the shorter signal.

To create a composite stereo press  on the toolbar of the main Sound Cleaner window. The program will display a dialog window (Figure 17.3).

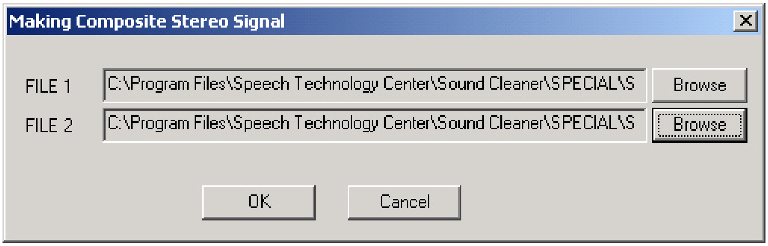


Figure 17.3: Composite stereo creation window

You will have to select files to be included in a composite stereo. If they meet all the necessary conditions, you may load the signal and process it as a common stereo file with all the available filters.

Chapter 18

Creating and editing processing schemes

During noise-cancellation you will once surely face tasks, unaccomplishable by standard means and typical schemes. That's why Sound Cleaner includes lots of possibilities for flexible tuning of processing procedure. One of these is creating you own unique **processing scheme** or editing an existing one.

Scheme window

Scheme window (Figure 18.1) displays and manages signal processing sequence. This window is usually located at the left edge of the main Sound Cleaner window. You may open it with **Project/View scheme** if the window is not displayed by default.

The scheme is displayed in the window as a sequence of icons. Each one represents a process included in the current scheme. Icons are connected with black lines, which show the way of a processed signal passing through the scheme. Icon's background color indicates status of the process: orange background marks active processes and gray – passive

Figure 18.1: Scheme window

ones. Double-clicking on an icon places its window on top in the Sound Cleaner window.

Icons of all the available processes are displayed in a column to the right of the scheme. There are also up and down arrows for scrolling the window. Note, that left-clicking on these arrows will scroll the scheme and right-clicking – available icons.

At the right edge of the window there is a column of **scheme control**

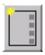
buttons. They correspond to commands of the system menu (you may open it by clicking over the icon in the window header):


Load scheme. Choose an existing scheme and load it.

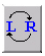

Save scheme. Save current scheme.


Change scale. View the scheme in smaller scale. It is useful for large and complicated schemes, which do not fit the regular window.


Redraw scheme. Automatically redraw the scheme optimizing its structure. Before you redraw the scheme, ensure, that there are incoming connections for all the processes (except sound input).


Erase scheme. Clears currently loaded scheme.


Change style. This button minimizes scheme window hiding all the icons and control buttons. To return to default window style activate system menu command.



Swap input channels. Swaps incoming channels for stereo signals.

Creating and adjusting the scheme

Current scheme is displayed in the scheme window. To add a new process find its icon in the column of available processes and drag it to desired location in the scheme. The program will automatically connect the process to its neighbors in the scheme. Do not forget, that there may be only one sound input and one output process in any scheme.

Clicking right mouse button over an icon in the scheme will open its context menu, which contains a list of available operations with scheme

elements:

Activate/De-Activate. These commands make the process active or passive.

Delete. Remove the process from current scheme.

Show/Hide/Minimize. Standard process window operations.

Create/Delete connection. After you select one of these commands, the cursor will change to . Then just click it over the icon to be connected or disconnected. This operations may be necessary if automatic connections were made in a wrong way.

Transpose inputs. Swap incoming channels for a selected process.

Duplication

Sometimes it is necessary to duplicate the signal and process it in two different modules independently. Most evident example is **saving** processed signal to file **playing** it at the same time to evaluate processing results. For such cases there is an additional **Duplicate** module. Being placed in a scheme, it receives a signal and then passes it to two different modules.

Typical schemes

Typical schemes included in Sound Cleaner demonstrate its performance when dealing with different types of noises and interferences. It may also help an unexperienced operator to master the program. As you choose the scheme, the program will automatically load preset filter settings and an example of signal to be processed. You may configure the program so, that no example will be loaded (see **Options** in **Basic principles**). You may also save your own schemes as typical.

Loading a typical scheme

To load a typical scheme choose **Load...** command from **Typical schemes** menu and then select a scheme. Each of the schemes in the list is provided with a short pop-up comment. It specifies, which type of noise or interference the scheme is designed to remove.

Saving current scheme as typical

To save your own typical scheme activate **Save as...** command in **Typical schemes** menu.

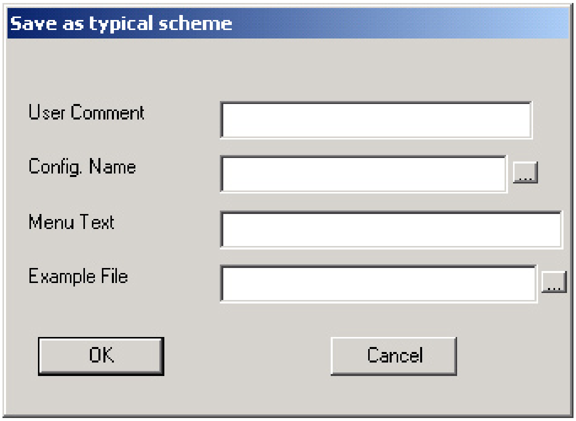


Figure 18.2: Saving typical scheme dialog window

In the dialog window(Figure 18.2) you will have to fill four scheme information fields:

User Comment – pop-up comment to a scheme.

Configuration Name – name of a file, where the scheme will be stored (without extension).

Menu Text – Name of the scheme, that will be displayed in the list of existing schemes.

Example file (optional) – sound example file, which will be loaded with the scheme.

Chapter 19

Sound Cleaner as Direct-X plug-in

Direct-X plug-in mode is supported by Sound Cleaner version 5.10 or higher. It does not affect any other program features. As a Direct-X plug-in Sound Cleaner is proved to be compatible with:

- Adobe Audition 1.0 or higher;
- Sound Forge 6.0 or higher;
- WaveLab 4.0[1]

Direct-X mode is activated by pressing . In this mode **Audio input** module will acquire sound data directly from a sound editor, while **Audio output** will pass the filtered signal back to the editor.

Main advantage of this mode is that Sound Cleaner may now get audio data from the sound editor, i.e. all the file types and formats supported by the editor may now be processed by Sound Cleaner.

[1] **Bypass** mode of WaveLab is not supported.

To run Sound Cleaner as Direct-X plug-in, please, follow these step-by-step instructions:

1. Start Sound Cleaner and enter Direct-X mode by pressing  on the toolbar.

2. Run your sound editor and activate **STC Sound Cleaner plug-in** using this editor's Direct-X plug-in activation procedure.

> ***Important!***
> If you are using Adobe Audition, you will have to select **Effects/Refresh Effects** before running Sound Cleaner plug-in.

3. Start playback of the fragment in the sound editor, then switch to Sound Cleaner and adjust processing scheme as necessary.

4. Return to sound editor window to stop the playback.

You should try to avoid running several sound editors at the same time, as it may lead to Sound Cleaner failures.

Chapter 20

Warranty

The developer guarantees, that the software conforms to the technical requirements, whereby the user observes the conditions and regulations of operation, storage and transport, for a period of **12 months** from the date of sale.

Tested and Approved

Sound Cleaner software, serial number ________ conforms to technical requirements and documentation and is declared suitable for use.

Adjustment conducted by ______________________

Date of issue ________________________________

Support

Our developers are always ready to assist you. In case of any questions, please don't hesitate to contact us:

Tel:	+7 (812) 331-0665
Fax:	+7 (812) 327-9297
E-mail:	info@speechpro.com
Web-site:	http://www.speechpro.com